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Reconfigurable Bandwidth Scheduler for multimedia Traffic in DiffServ Router

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Abstract: Quality of service (QoS) provisioning, in today's IP networks, is becoming more important due to the tremendous growth of real-time and multimedia applications. However, one of the challenges to achieving QoS requirements is how to allocate the bandwidth to various applications. One solution is to assigning bandwidth dynamically instead statically. In this paper, a reconfigurable bandwidth allocation technique based on QoS provisioning over DiffServ router is proposed. The technique aims at improving the performance and bandwidth utilization of multimedia applications by allowing video traffic class to share the bandwidth of other traffic classes that can tolerate a certain percentage of packet loss without any degradation of quality. This property is used to adjust the bandwidth assigned to video traffic according to the percentage of the packet loss of other classes. As a study case, the reconfigurable bandwidth allocation scheme is applied to a network model based on a modified deficit round robin (MDRR) scheduler. It can be seen from the simulation, using OPNET modeler 14.5, that the proposed QoS model improves the QoS of multimedia traffic.

Keywords: Reconfigurable Bandwidth, QoS, DiffServ Router, MDRR.

1. Introduction

Quality of service offered by any network depends on many factors such as available bandwidth, delay, jitter, packet loss etc.[1]. Multimedia traffic usually requires considerable amounts of bandwidth in order to provide acceptable QoS to the end-user. However, bandwidth resources are always limited and a scarce resource; even in networks with sufficient bandwidth an "insurance policy" is essential to ensure guaranteed quality for real-time applications, regardless of the overall network traffic load. Thus, a service must extract the maximum profit benefit from every bit of bandwidth available.

Current IP networks can benefit from an intelligent mechanism (such as reconfigurable technique) that dynamically adjusts bandwidth allocated between traffic classes in the network to accommodate the multimedia traffic class. In Differentiated Services (DiffServ) networks, the issue of bandwidth allocated has attracted a lot of attention from researchers. Due to the interest in providing QoS in IP networks with the use of DiffServ, we are motivated to use the DiffServ architecture in the design of a reconfigurable bandwidth scheduler.

This paper introduces a new method for reconfigurable distribution of available bandwidth in DiffServ router among traffic classes based on packet loss. The scheduling policy used is based on a modified deficit round robin (MDRR) algorithm. Some applications can tolerate a certain percentage of packet loss without any degradation of quality. Based on this fact, we used drop percentage ratio (or packet loss rate), to reconfigure (redistribute) the available bandwidth between the traffic classes to give preference to the real-time traffic class over other traffic classes if the packet loss rate of these traffic classes are less than the allowable rate, according to QoS parameters.

The paper is organized as follows: in section 2, we give a brief overview of previous work in the field. Quality of service parameters is outlined in section 3. Section 4 and 5 present a brief description of Modified Deficit Round Robin scheduling algorithm and bandwidth allocation technique, respectively. The architecture of the proposed system is depicted in section 6. Finally, in section 7 and 8 we present the simulation results and conclusion of the work, respectively.

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2. PREVIOUS WORKS

There is active ongoing research to find the mechanisms that would be able to provide dynamic bandwidth management in networks with a variety of communication media and devices while providing the required QoS level for services.

In [2], the authors proposed a dynamic resource allocation scheme to support the constantly increasing online video stream traffic, especially high definition video streams based on online traffic prediction using seasonal time analysis. The proposed scheme is able to predict the required bandwidth for future video frames depending only on the information available from the incoming video stream (content-based) by renegotiating the assigned bandwidth for that flow.

X. Li *et al.* [3] have studied the problem of a class-based bandwidth allocation and admission control with DiffServ. They first presented two basic approaches: dynamic bandwidth allocation with parameter-based admission control, and static bandwidth allocation with parameter-based admission control. They proposed the framework of an intra-class bandwidth allocation and an inter-class request admission control to support QoS VPN provisioning at the edge of core networks.

To enhance QoS of real-time traffic, Hussain *et al*.[4] propose a dynamic bandwidth allocation scheme that alleviates congestion problems in gateway-based multi-hop WiFi-based long distance networks. In this research, the unused bandwidth is distributed among the needy nodes by using a dynamic slot scheduling scheme. Wang *et al*.[5] propose a bandwidth estimation algorithm using a top-down scheme that combines the features of one-way delay (OWD) and packet dispersion focusing on TCP-friendly congestion control of fair end-to-end video streaming by inferring available bandwidth.

They considered the characteristics of streaming applications, such as the bandwidth resolution in scalable video coding (SVC) which can achieve fine granularity of scalability at the bit level to fit the time-vary heterogeneous networks. Also, they presented a congestion control algorithm for SVC based streaming through the tool of bandwidth inference by periodical probing.

The main objective of Baghla *et al*.[1] work is to improve QoS in the network by reducing bandwidth consumption and link load. A mathematical model is designed for the efficient use of the bandwidth required by an MPLS routing path. In this work, the effect of parameters, like packet loss, transmission delay at each link etc., on the effective bandwidth required by any transmission path is also considered.

The article of Tsang *et al*.[6] addresses the challenges of the dynamic bandwidth allocation (DBA), where increased reach, in order to achieve a larger degree of node consolidation, results in a degradation of DBA performance and quality of service support. The Real Time Probabilistic Systems is used to evaluate atypical PON systems performance. He analyzes how changes in performance depending on changes in particular modes by supplying ranges for parameter values. And he compares the proposed algorithm with traditional DBA and shows its advantage on average packet delay.

In a research article of T. Balogh *et al*.[7], an iterative method is proposed for the calculation of average bandwidth assignment to traffic flows using a WFQ scheduler in IP based NGN networks. The bandwidth assignment calculation is based on the link speed, assigned weights, arrival rate, and average packet length or input rate of the traffic flows. Their model is then used for the analysis of the impact of weight settings, analyzing the stability of the system and modeling of delay and queue length of traffic classes.

A cost-efficient bandwidth provisioning and reconfiguration, called Capacity Allocation using Time Zones is proposed by Cavdar *et al*. [8]. Different bandwidth allocation methods have been studied for setting up a worldwide layer 1 VPN. They evaluated the state of the art in layer 1 VPNs in the context of globally deployable optical networks and cost-efficient dynamic bandwidth usage.

3. QUALITY OF SERVICE PARAMETERS

The important QoS parameters for efficient performance of a network are[9]:

a. End-to-end delay

The end-to-end delay consists of the following:

- The ***transmission delay*** at a node
- The ***propagation delay*** on the link to the next node,
- The ***processing*** and ***queueing delays*** at the node.

Propagation and queuing delay are the key contributors to delay as long as no heavy processing like encryption or packetization by applications is needed.

b. Delay jitter (Delay variation)

The delay jitter is defined as the absolute difference of the delays between received packets.

c. Packet loss

The packet loss rate indicates the number of packets that do not reach the destination in relation to all transferred packets. There are three sources of packet loss in an IP network: a break in a physical link, noise and network congestion that leads to buffer overflow.

The organizations 3GPP and TISPAN (Telecoms & Internet Converged Services & Protocols for Advanced Networks) define a set of four classes of QoS for transport networks. Corresponding to the IMS services, these requirements also apply to the interconnection of networks that are IP-based (IP-CAN) access networks and IMS [10]. These four classes are listed in Table 1.

TABLE 1. QOS CLASSES FOR TRANSPORT NETWORKS [11]

Class	Key Attributes	Utilization
Conversational	Responsive to delay variation, limited tolerance to packet loss	Audio/Video conversation
Streaming	Responsive to but tolerant of delay variation, limited tolerance to packet loss	Video streaming
Interactive	Responsive to round-trip delay, packets transferred transparently with low bit error rate.	Collaboration, conference
Background	Insensitive to delay transparently with low bit error rate.	Email, IM, chat

4. MODIFIED DEFICIT ROUND ROBIN ALGORITHM

In order to achieve the required performance guarantees for a real-time application such as voice and video, a Quality of Service (QoS) provisioning is needed in multi-service packet networks. The scheduling algorithm is a key component for QoS enabling networks, that determines the order in which packets are dequeued to the interface for transmission on the physical wire. Different scheduling algorithms have been devised during the last two decades, with different fairness and latency properties. Among those, Deficit Round Robin (DRR) [12] which is an extension of Weighted Round-Robin (WRR) that allows flows with variable packet lengths to share the link bandwidth fairly. It is used in high-speed implementations because of its low complexity and ease of implementation. However, due to its round robin structure, its latency properties are not adequate for real-time applications.

To solve this problem variants of the basic DRR algorithm that prioritize one (or a subset of) queue(s), so as to guarantee a reduced latency named as Modified Deficit Round Robin (MDRR) and implemented in commercial routers[13]. The MDRR scheduler provides relative bandwidth guarantees and allows the definition of a *low latency queue (LLQ)*, which is given priority over the set of standard queues and is mainly intended to support voice traffic. Other standard queues are serviced according to DRR. When a queue is served, MDRR keeps track of the number of bytes of data that was dequeued in excess of the configured value. In the next pass, when the queue is served again, fewer data will be dequeued to compensate for the excess data that was served previously. As a result, the average amount of data dequeued per queue will be close to the configured value.

Each queue within MDRR is characterized by two variables:

Quantum value – Hold the average number of bytes served in each round.

Deficit counter – used to track how many bytes a queue has transmitted in each round. It is initialized to the quantum value. Packets in a queue are served as long as the deficit counter is greater than zero. Each packet served decreases the deficit counter by a value equal to its length in bytes. A queue can no longer be served after the deficit counter becomes zero or negative. The deficit counter of each non- empty queue is increased by its quantum value in each new round. Each MDRR queue is given a relative weight which assigns relative bandwidth to that queue when the interface is congested.

The LLQ in MDRR can work either in *strict* or *alternate* priority mode. In the strict priority mode, the LLQ is *always* serviced in an exhaustive, non-preemptive priority mode. The other queues are serviced cyclically, as in DRR, whenever the LLQ is empty. A standard queue can thus have its service turn interrupted by the arrival of a packet in the LLQ. The behavior of strict priority MDRR is illustrated in Figure 1. In an alternate mode, the scheduler completely serves the non-priority queue until its deficit counter reaches zero. The PQ during this time is not serviced, and the VoIP packets are delayed [14].

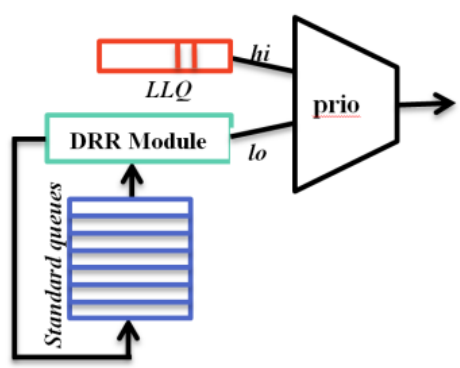


Figure 1. – strict-priority in MDRR[15]

5. DYNAMIC BANDWIDTH ALLOCATION

The proposed reconfigurable bandwidth allocation scheme aims to improve the performance of the real-time applications in DiffServ router. Our solution to achieve this goal is by assigning extra bandwidth to video traffic from the bandwidth of the other traffic classes(classes 0-5 except the voice traffic class) according to their packet loss conditions. Compared to static allocation, this method has a major advantage that improves both bandwidth utilization and the performance of the video traffic applications. The bandwidth of the video traffic class (class 6) is shared with the other traffic classes (traffic classes 0-5) as long as the packets loss percentage of the traffic classes (0-5) within the acceptable level that achieve good quality of traffic class at the destination side and if the queue length of video traffic is full.

However, it is equally important not to disturb other traffic classes so that fairness is assured to other traffics. This is achieved by allowing class 6 traffic to share the bandwidth with other traffic classes only if the packet loss ratio within the acceptable level as recommended by the IETF which assumed 1% for all traffic classes from class 0 to class 5. Figure 2 shows a flow chart illustrating the reconfiguration code in the RB-MDRR scheduler, where:

-Prec is the precedence field in the packet header which is used to classify the arrival packets.

-Next_Queue is the next queue to be serviced after it is selected by the MDRR or PQ scheduler.

-Drop_percent is the packets loss ratio.

Packet loss is the difference between the generated and received packets. The packet loss ratio is defined as follows[16]:

$$\textit{packet loss ratio} = \frac{\text{total number of packets dropped}}{\text{total number of packets received}} \times 100\% \quad (1)$$

The packet loss ratio shows the throughput of the scheduling scheme. A lower packet loss ratio indicates a better throughput.

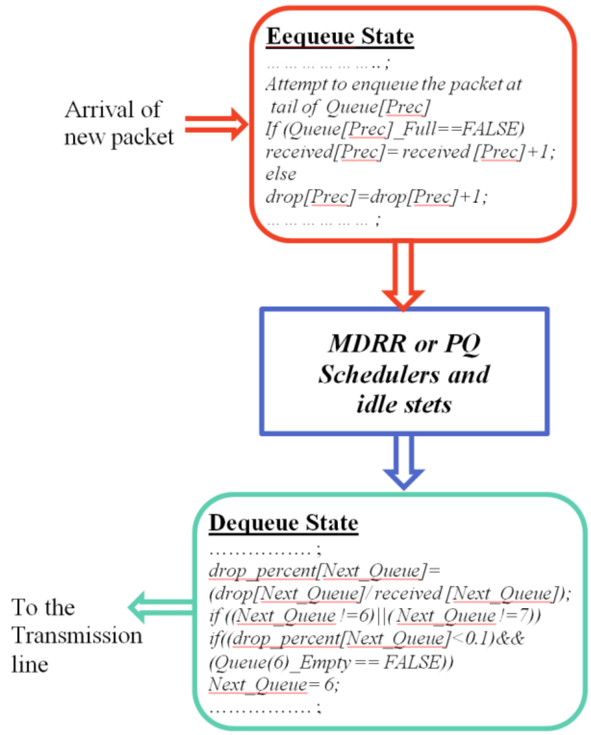


Figure 2. The Pseudocode of the reconfiguration process in the RB MDRR scheduler

6. SYSTEM ARCHITECTURE

Figure 3 shows the network topology used in OPNET software to simulate the Reconfigurable Bandwidth MDRR scheduler model (**RB-MDRR**). The link between the scheduler and server is the only bottleneck link with a service rate of 1M bit/sec. All Other links have enough bandwidth. The offered traffic load that goes to the bottleneck link of the network is equal to the sum of all aggregated traffic at the queues.

A "single bottleneck" topology is implemented to compare the performance of different schedulers on a single router. There are 8 source nodes generating traffic to the server each source node consists of four traffic generators activated at different simulation times.

In our simulations, all the packets have the same size (500bit), as an ATM cell. Therefore, the bottleneck link of 1Mbit/sec has a speed of sending 2000 packets per second. Since the purpose of the simulations is to compare the performances of different schedulers, the absolute values of link speed or packet size or traffic sources rates will not affect our results as long as they are set to the same for all schedulers[17].

6.1 RB-MDRR scheduler model assumptions

The performance of the RB-MDRR scheduler will be compared with the performance of Cisco MDRR. The assumptions to be used during simulation of the two schedulers are as follows:

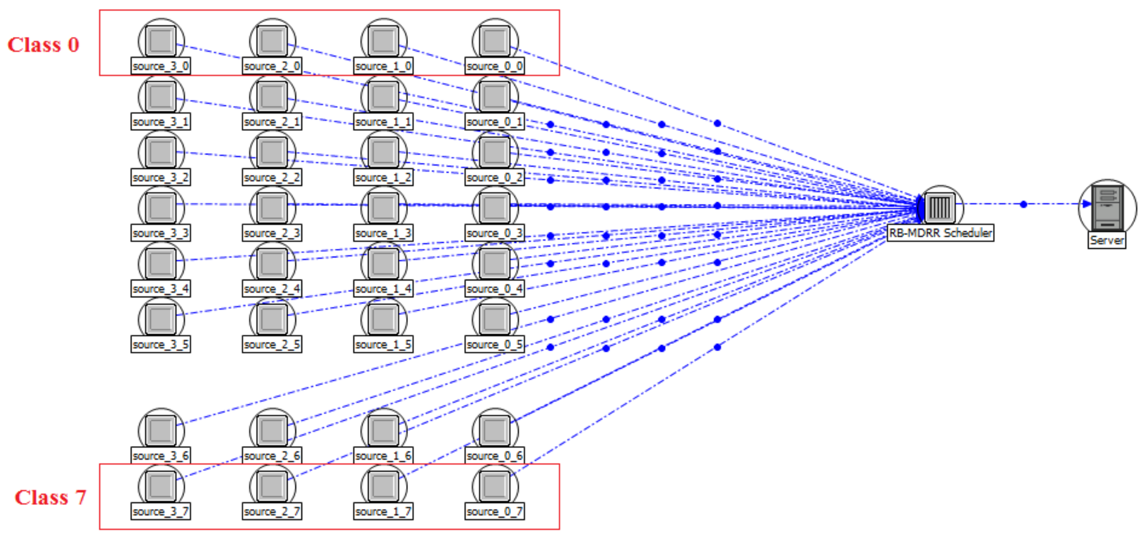


Figure 3. Network topology

1) The voice traffic is represented by four Constant Bit Rate (CBR) traffic sources with a constant packet size of 500bit.
2) Packets of traffic classes (0-6) get served only if there are no EF packets waiting.
3) The video traffic is represented by four Variable Bit Rate (VBR) traffic sources.
4) The video frame is segmented into equal size packets with a size of 500 bits.
5) The packets size of all other traffic classes is equal to 500 bit.
6) The bottleneck link has a speed of sending 2000 packet per second (1Mb/s).
7) The buffer space for all traffic classes is fixed at 500 packets.
8) The distribution of inter-arrival time in all traffic sources are exponentially distributed with different mean values.
9) The traffic in all classes is assumed to be Poisson distributed.
10) All the queues use the Tail-drop queue management scheme.
11) An eight traffic classes approach is adopted in the implemented scheduler, as recommended by Cisco for DiffServ, illustrated in Table 4 [18].
12) The simulation time is 30 minutes.
13) The four steps of the traffic conditions (depending on the assigned weight of each traffic class, see Table 4) of the classes (0-7) are: step 1 is under-provisioning, step 2 is provisioning and step 3 and 4 are over-provisioning. These steps correspond to 50%, 100%, 150%, and 200% of a full load of output link respectively (see Table 2).
14) In order to study the performance of the proposed reconfigurable scheduler at different traffic conditions, the traffic sources are activated at different times (in the four steps explained above) as shown in Table 3.

Figure 4 shows the state transition diagram of the RB-MDRR scheduler. It consists of seven states as follows:

- *The stats init_1, init_2, are* used to initialize the variables, constants, statistics and to read subqueues attributes.
- *enqueue*, and *dequeue* used to enqueue and dequeue the packet to and from appropriate queue.
- *Sched_MDRR and Sched_PQ* implement the scheduling algorithms MDRR and PQ respectively.
- *idle state*.

All states are forced states except the *idle* state which is an unforced state. In the simulation, the delay, and packet loss of each subqueue (using built-in statistic functions in OPNET) are measured.

TABLE 2 TRAFFIC CONDITIONS (BIT/SEC) FOR CLASSES 0-7

Traffic condition	Step No.	Class 0	Class 1	Class 2	Class 3	Class 4	Class 5	Class 6	Class 7
Under-Provisioning	1	5000	125 000	120 000	50 000	10 000	25 000	115 000	25 000
provisioning	2	10 000	250 000	240 000	100 000	20 000	50 000	230 000	50 000
Over-provisioning	3	15 000	375 000	360 000	150 000	30 000	75 000	345 000	75 000
	4	20 000	500 000	480 000	200 000	40 000	100 000	460 000	100 000

TABLE 3 PARAMETERS SETTING FOR TRAFFIC CLASSES 0-7

Traffic Class	0 - 7			
Step No.	1	2	3	4
Activated Traffic sources	Source_0-0 to Source_0-7	Source_1-0 to Source_1-7	Source_2-0 to Source_2-7	Source_3-0 to Source_3-7
Start Time(sec)	10	200	600	1200
End Time(sec)	1800	1800	1800	1800
Traffic percentage (of the full load)	50%	100%	150%	200%

TABLE 4 WAN QoS Class Models

Weight(%) / Class	4-Class Model	8-Class Model	12-Class Model
10 (class-7)	Realtime EF/CS4/CS5	Voice-EF	Voice-EF
23 (class-6)		Interactive Video-CS4/CS5	Broadcast Video-CS5
			Real-time Interactive-CS4
5 (class-5)	Signaling / Control-CS2/CS3/CS6	Network Control and Management –CS2/CS6	Network Control-CS6
			Network Management –CS2
2 (class-4)		Call Signaling-CS3	Call Signaling-CS3
10 (class-3)	Critical Data AF1/AF2/AF3/AF4	Streaming Video-AF3	Multimedia Streaming-AF3
24 (class-2)		Critical Data-AF1/AF2/AF4	Multimedia Conferencing-AF4
			Transactional Data-AF2
			Bulk Data-AF1
25 (class-1)	Best Effort	Best Effort	Best Effort
1 (class-0)	Scavenger	Scavenger	Scavenger

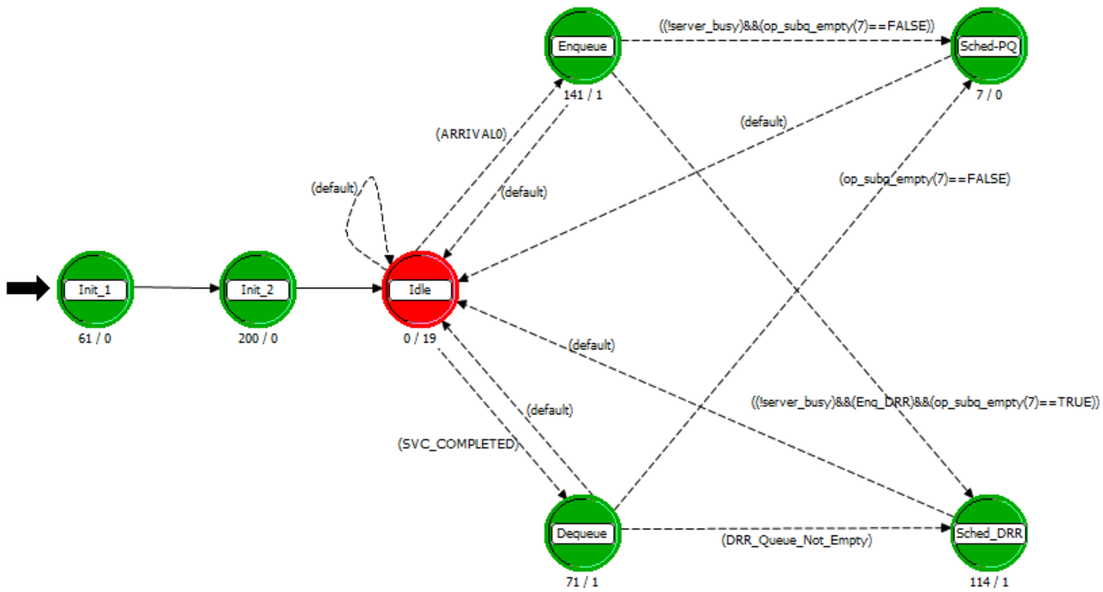
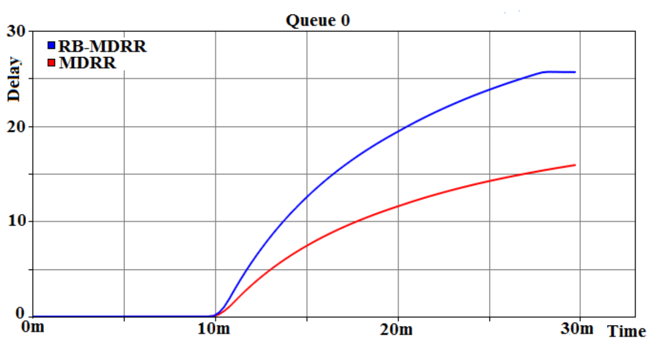


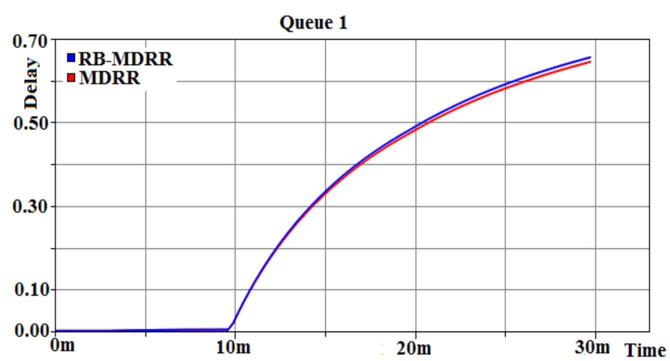
Figure 4 OPNET state transition diagram of the RB-MDRR Scheduler

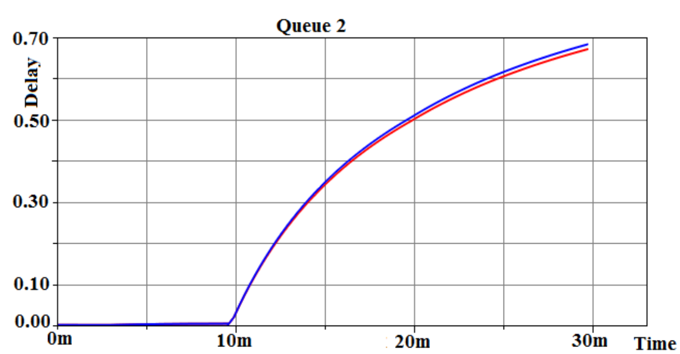
7. Simulation Results and Analysis

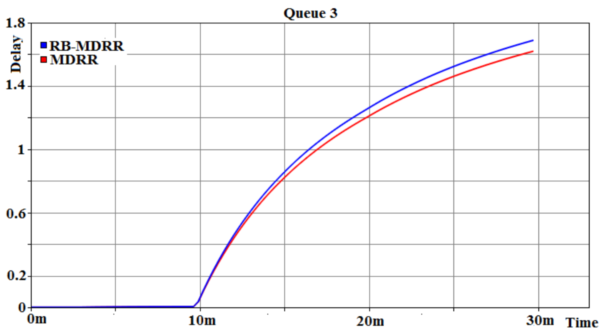
OPNET Modeler 14.5 is used to simulate the RB-MDRR model described previously. The performance of RB-MDRR will be compared with the performance of Cisco MDRR. As explained previously, to be fair, the input traffic, the buffer allocations and queue management schemes are set to the same under the two schedulers. The voice traffic, class 7, (implemented as EF class) has a strict priority mode and is not affected by the reconfiguration process.

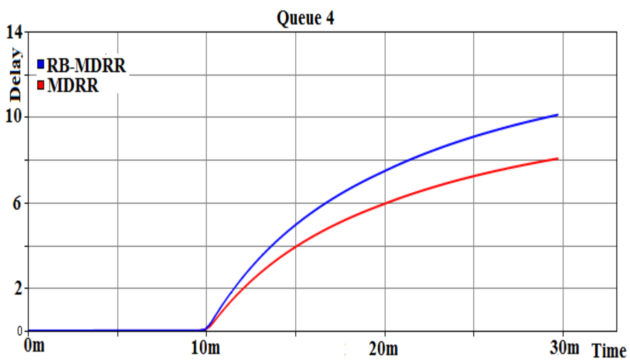
Figure 5 shows queueing delay (in ***seconds***) of traffic classes (0-7) versus simulation time (in ***minutes***) under RB-MDRR and MDRR schedulers. It is seen from

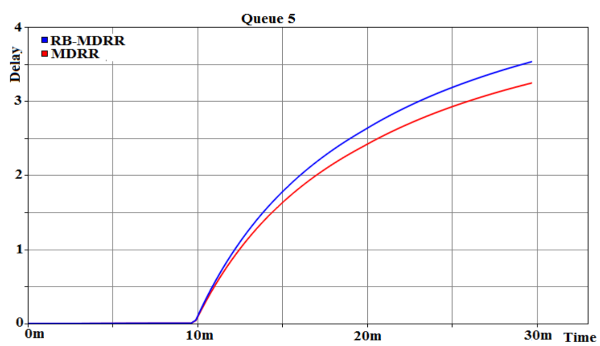


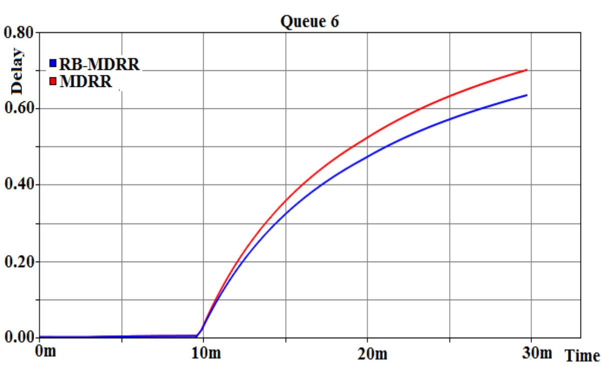












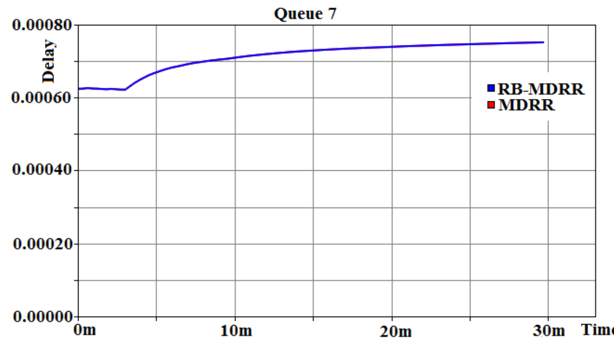
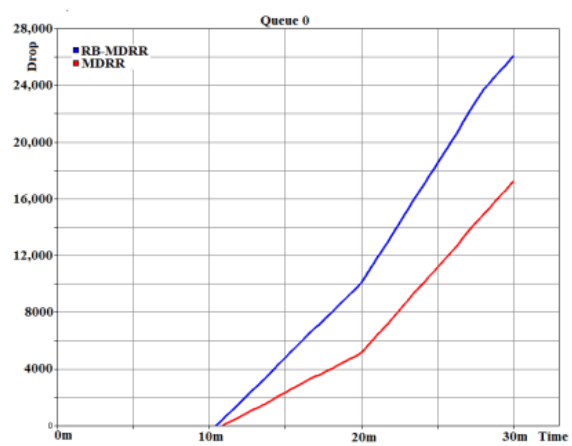
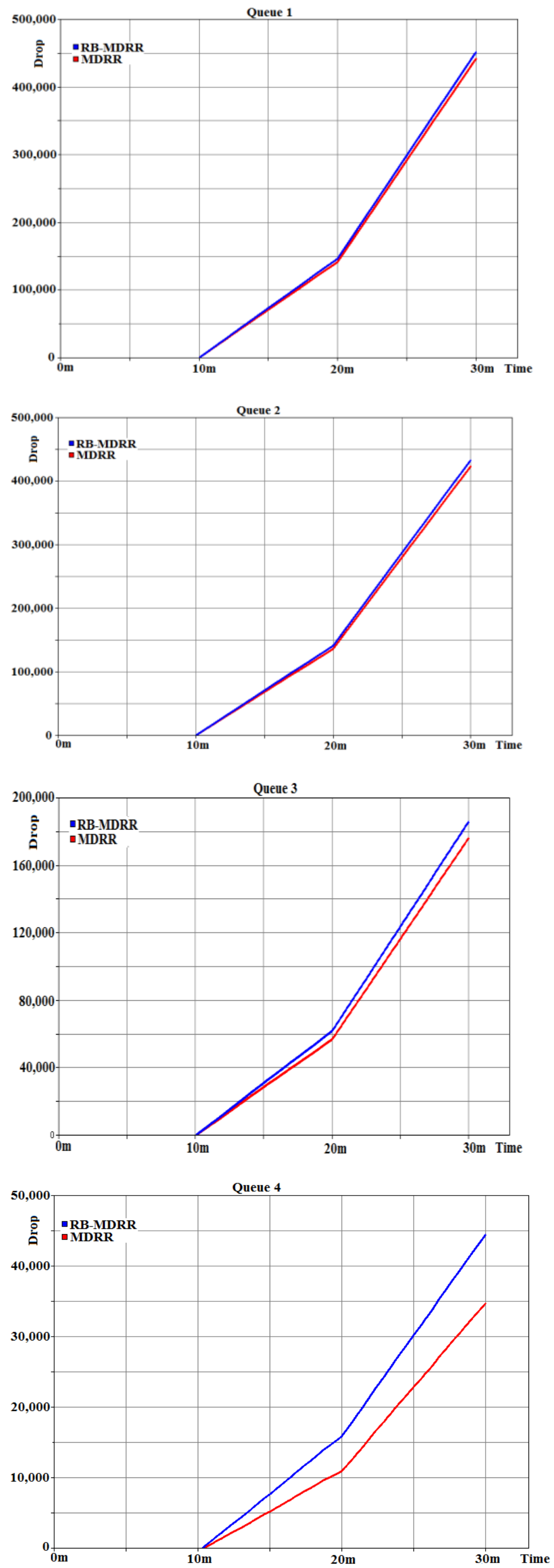


Figure 5. Average queuing delay of traffic classes 0-7 under RB-MDRR and MDRR schedulers

Figure 5 that the RB-MDRR scheduler improves the QoS of the traffic class 6 (video traffic) by reducing the queueing delay at the cost of increasing the delay of other traffic classes (0-5). However; packets in traffic classes (0-5) continue to get good QoS as long as the packet loss ratio does not exceed the recommended limit. The percentage of increase in the queue delay of a traffic class is proportional to the weight assigned to that class.

Simulation results of packets drop under RB-MDRR and MDRR schedulers for various traffic classes are presented in Figure 6.





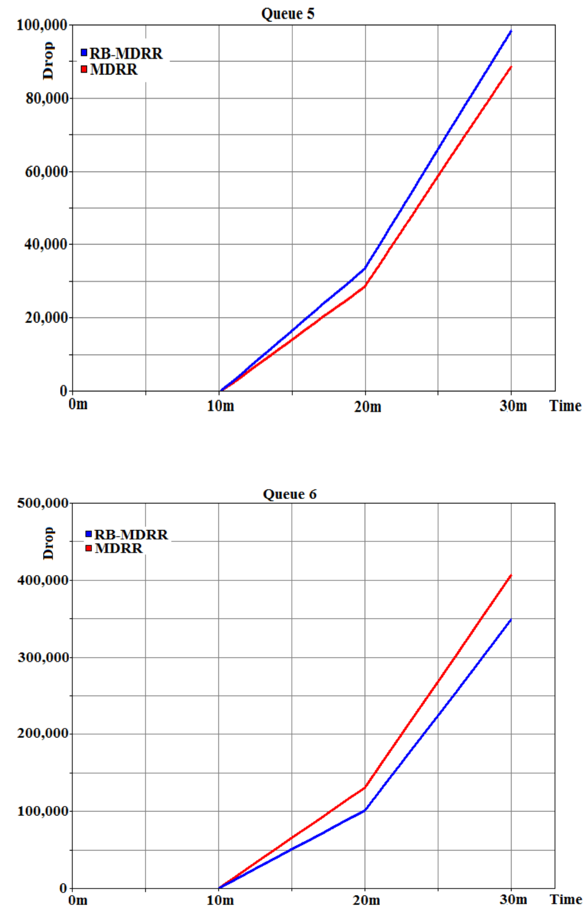


Figure 6. Packet dropped in traffic classes 0-6 under RB-MDRR and MDRR schedulers

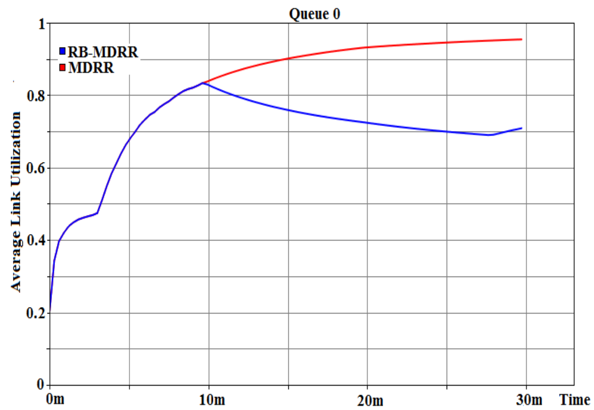
The figure shows that the packet loss at video traffic (class 6) in RB-MDRR is decreased, though the total offered load at the bottleneck link increases beyond the available (constrained) bandwidth at the bottleneck link. This is due to the extra bandwidth that is added to video traffic.

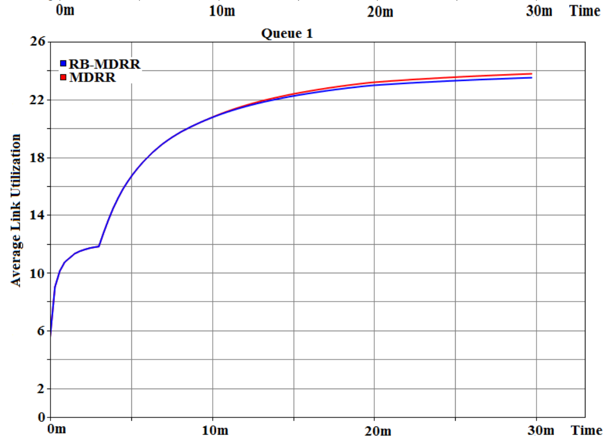
The packet loss at traffic classes (0-5) under RB-MDRR is increased, compared to the packet loss under the MDRR scheduler, as expected, with the increasing of the total offered load at the bottleneck link. This is because traffic classes (0-5) get downgraded when adaptive bandwidth allocation is employed. However, the packet loss of traffic classes (0-5) is still within the acceptable level which ensures attaining good QoS. Again, the percentage of increase in the packet loss of traffic class (0-5) is proportional to the weight assigned to that class.

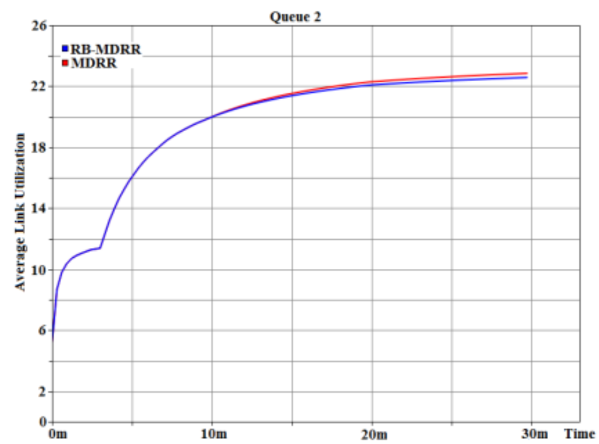
TABLE 5. PACKET DROPPED UNDER RB-MDRR AND MDRR AT SIMULATION TIME 1800 SECOND

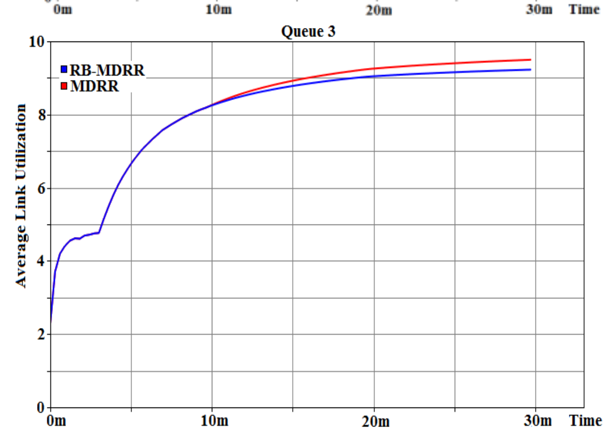
Queue No.	Packet drop under RB-MDRR at simulation Time 1800 sec	Packet drop under MDRR at simulation Time 1800 sec	Difference
0	26053	17223	8830
1	451174	441440	9734
2	432809	423076	9733
3	185487	175754	9733
4	44393	34660	9733
5	98331	88598	9733
Sum of differences in traffic classes 0-5			57496
6	348482	405981	-57499
7	-	-	-

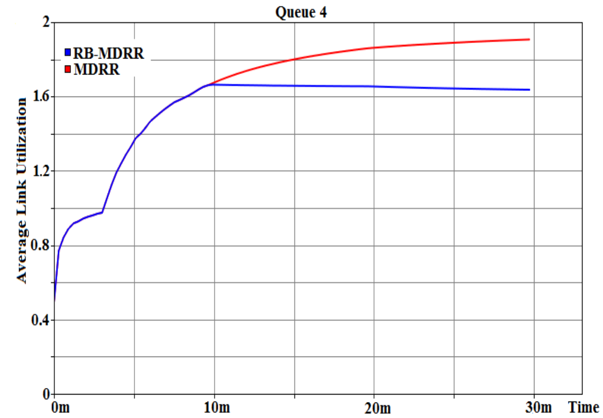
Since we assume that packet size on all traffic classes are the same and all traffic classes have the same distribution, at any simulation time, the decreasing rate of the packet loss of video traffic is equal to the sum of the rates at which the packet loss of traffic classes (0-5) is increased. For example, Table 5 illustrates the differences in packet dropped under RB-MDRR and MDRR, at simulation time 1800 second.

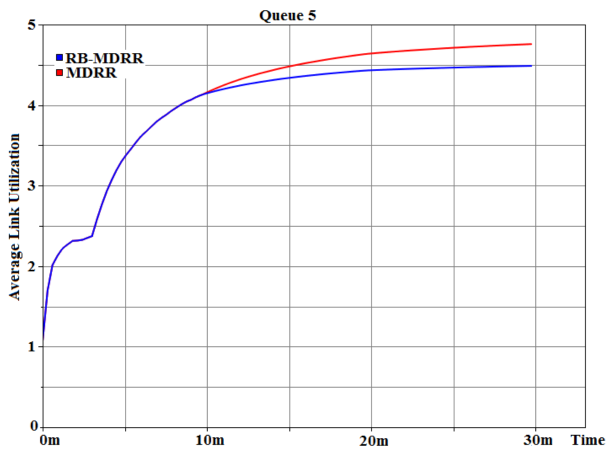


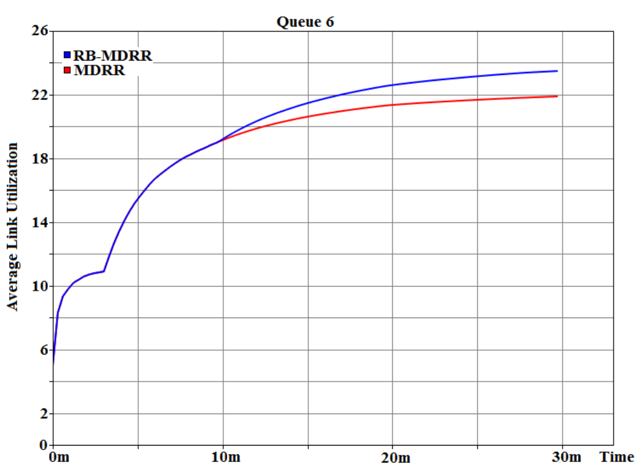












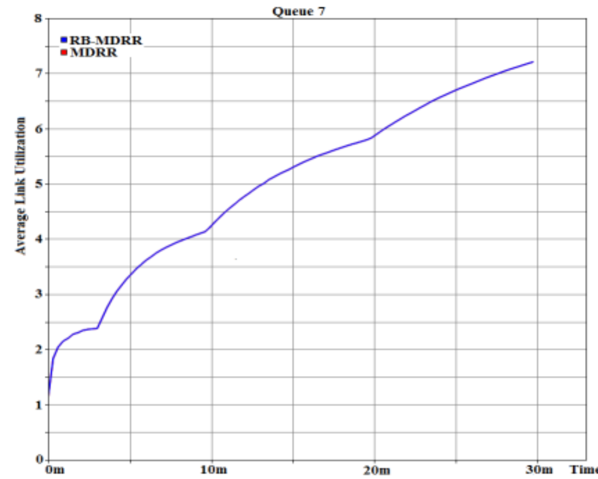


Figure 7. Average link utilization of traffic classes (0-7)

The extra bandwidth that is added to video traffic also improves the link utilization of video traffic as shown in figure 7. Link utilization is defined as the ratio between the amount of data carried on the link and the link's capacity. These statistics record the link utilization as a value between 0% and 100%, where 100% means that the link's capacity is fully consumed.

It is clear in figures 5, 6, and 7, that the proposed RB-MDRR and standard MDRR provide similar results when the there is no congestion (i.e. the traffic load is equal or less than 100% of the full load). This occurs at the simulation time of less than 10 minutes.

As in queuing delay and packet drop, the degradation in the link utilization of every traffic class is proportional to the weight of that traffic class. For the traffic classes 1, 2, and 3 which have weights of 25%, 24%, and 10%, respectively, we see that there is a small degradation in link utilization. However, the link utilization of video traffic (class 6) is increased beyond the assigned weight when the total offered load at the bottleneck link increased beyond the initially assigned bandwidth. So, the bandwidth utilization of voice and video is sustained during the reconfiguration process. By this method, the bandwidth adjustment is kept up with the QoS conditions, leading to providing an improvement in QoS of multimedia applications.

8. Conclusion

In this paper, a reconfigurable bandwidth architecture implemented and deployed within MDRR scheduler, namely RB-MDRR, is proposed. The technique aims at improving the performance of video traffic applications by allowing the video traffic class to share the bandwidth with other classes (except the voice traffic) if the QoS of these traffic classes within the recommended levels. In this dynamic approach, the QoS measured based on the percentage of packet loss and then the bandwidth allocations of video traffic class, are adjusted according to the packet loss conditions of other classes. If traffic classes (0-5) currently receive QoS within the acceptable level, and the video traffic class has bad QoS, then the bandwidth allocated to traffic classes (0-5) will decrease to allow the video traffic class to use the extra bandwidth.

The service providers always want to achieve a high utilization, approaching 1.0, but need to maintain QoS commitments. Based on the proposed reconfigurable BW scheduler, a service provider can maintain the highest load for video traffic under which the delay and loss of other traffic classes are acceptable.

We have analyzed the performance (in terms of queueing delay, packet loss, and link utilization) of the proposed scheduler and compared with MDRR scheduler, implemented on Cisco 12000 router using OPNET simulation. It has been seen from the results of a simulation that the proposed reconfigurable bandwidth allocation model improves the performance of video traffic, and at the same time preserves good QoS of other traffic classes. According to our reconfigurable scheme, if the video traffic receives a bad QoS then it gets priority over other traffic (except voice traffic). Finally, the proposed reconfigurable bandwidth architecture can be applied to other schedulers such as CBWFQ.

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